Cisco − VoIP with PPP

Table of Contents

Voice over IP Quality of Service for Low−Speed PPP Links (IP RTP Priority, LFI, cRTP)

Contents

Introduction

This sample configuration studies a VoIP with Point to Point Protocol (PPP) over low bandwidth leased line configuration. The document includes background technical information on the configured features, design guidelines, and basic verification and troubleshooting strategies.

It is important to note that in the configuration below, the two routers are connected back−to−back over a leased line. In most topologies however, the voice enabled routers can exist anywhere. Usually, the voice routers use LAN connectivity to other routers which are connected to the WAN (i.e. PPP leased line). This is important because if your voice routers are not directly connected via PPP over a leased line, all WAN configuration commands must be configured on those routers connected to the WAN, and not on the voice routers, as shown in the configurations below.

QoS Design Guidelines for VoIP Over Low−Speed PPP Links

This section provides design guidelines to configure Voice over IP over a low−speed PPP leased line. To ensure voice quality Cisco provides a rich set of QoS features that allow you to classify data and voice traffic into different categories and satisfy the delivery of real−time voice packets.

Link Fragmentation and Interleaving (LFI): Multilink PPP

Large data packets can adversely delay delivery of small voice packets, reducing speech quality. Fragmenting these large data packets into smaller ones and interleaving voice packets among the fragments reduces jitter and delay. The Cisco IOS® Link Fragmentation and Interleaving (LFI) feature helps satisfy the real−time delivery requirements of VoIP.

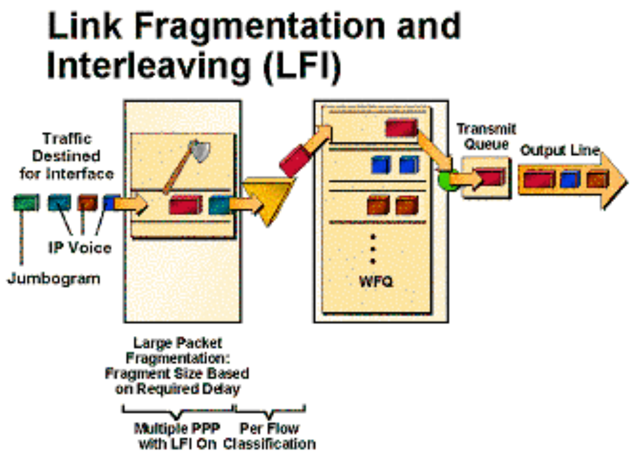
As shown in Table 1, the amount of serialization delay (the time it takes to actually place the bits onto an interface) introduced on low−speed WAN links can be significant, considering that the target end−to−end one−way delay should not exceed 150ms. (ITU−T G.114 recommendation specifies 150ms maximum one−way end−to−end delay for high voice quality). For voice applications, serialization delay should not exceed 20 ms.

Serialization Delay = frame size (bits) / link bandwidth (bps)

	1 Byte	**64 Bytes**	**128 Bytes**	**256 Bytes**	**512 Bytes**	**1024 Bytes**	**1500 Bytes**
56 kbps	143 us	9 ms	18 ms	36 ms	72 ms	144 ms	214 ms
64 kbps	125 us	8 ms	16 ms	32 ms	64 ms	126 ms	187 ms
128 kbps	62.5 us	4 ms	8 ms	16 ms	32 ms	64 ms	93 ms
256 kbps	31 us	2 ms	4 ms	8 ms	16 ms	32 ms	46 ms
512 kbps	15.5 us	1 ms	2 ms	4 ms	8 ms	16 ms	32 ms
768 kbps	10 us	640 us	1.28 ms	2.56 ms	5.12 ms	10.24 ms	15 ms
1536 kbps	5 us	320 us	640 us	1.28 ms	2.56 ms	5.12 ms	7.5 ms

Table 1. Serialization Delay for Various Frame Sizes on Low−Speed Links

The link fragment size is configurable in millisecond time measurements with the command **ppp multilink fragment−delay**. LFI requires that **ppp multilink** be configured on the interface with **ppp multilink interleave** turned on. For more information on configuring LFI, refer to the Sample Configuration section. The following image illustrates the operation of LFI.



Note: In cases where you have a dedicated full T1 connection, you may not need a fragmentation feature. (You will, however, still need a QoS mechanism such as IP RTP Priority, in this case). The full T1 offers enough bandwidth to allow voice packets to enter and leave the queue without delay issues. Also, you may not need Compression for Real−time Protocol (CRTP), which helps conserve bandwidth by compressing IP RTP headers, in the case of a full T1.

IP RTP Priority (also called PQ/WFQ)

IP RTP Priority (also known as PQ/WFQ: Priority Queue/Weigthed Fair Queuing) creates a strict−priority queue for a set of RTP packet flows belonging to a range of UDP destination ports. While the actual ports used are dynamically negotiated between end−devices or gateways, all Cisco VoIP products utilize the same UDP port range (16384−32767). Once the router recognizes the VoIP traffic, it places it into the strict priority queue. When the priority queue is empty, the other queues are processed according to standard Weighted Fair Queing (WFQ). IP RTP Priority does not become active until there is congestion on the interface.

To configure IP RTP Priority use the following guidelines:

- `Router(config-if)#`**`ip rtp priority`** *starting−rtp−port−# port−#−range bandwidth*

starting−rtp−port−number	Lower bound of UDP port. The lowest port number to which the packets are sent. **For VoIP set this value to 16384**.
port−number−range	The range of UDP destination ports. A number, which added to the *starting−rtp−port−number*, yields the highest UDP port number. **For VoIP set this value to 16383 (32767 − 16384 = 16383)**
bandwidth	Maximum allowed bandwidth (kbps) in the priority queue. Set this number according to the number of simultaneous

	calls the system will support.

For more information on bandwidth per call utilization refer to: Voice over IP − Per Call Bandwidth Consumption.

Compression Real−Time Protocol (cRTP)

The RTP header compression feature compresses the IP/UDP/RTP header from 40 Bytes to 2 or 4 bytes, reducing unnecessary bandwidth consumption. It is a hop−by−hop compression scheme, therefore cRTP must be configured on both ends of the link (unless the **passive** option is configured). To configure cRTP use the following command at interface level:

- `Router(config−if)#`**`ip rtp header−compression`** [passive]

The compression process can be CPU−intensive. Cisco only recommends using cRTP with links lower than 768 kpbs, unless the router is running at a low CPU utilization rate. Monitor the router's CPU utilization and disable cRTP if it's above 75%.

Note: In the configurations below, when you configure the command **ip rtp header−compression** , the router adds the command **ip tcp header−compression** to the configuration by default.

For more information: Compressed Real−Time Transport Protocol

Compress TCP Packet Headers

- `Router(config−if)#`**`ip tcp header−compression`** [passive]

You can compress the headers of your TCP/IP packets to reduce their size, thereby increasing performance. Header compression is particularly useful on networks with a large percentage of small packets, such as those supporting many Telnet connections. This feature only compresses the TCP header, so it has no effect on UDP packets or other protocol headers. The TCP header compression technique, described fully in RFC 1144, is supported on serial lines using HDLC or PPP encapsulation. You must enable compression on both ends of a serial connection.

Other Bandwidth Reduction Tips

- Use low bit−rate coder/decoders (codec) on the VoIP call legs;
  G.729 (8 Kbps) is recommended. (This is the default codec on the VoIP dial−peers). To configure different codecs use the command **router(config−dial−peer)#codec** under the desired voip dial−peer.

- Although dual tone multifrequency (DTMF) is usually transported accurately when using high−bit−rate voice codecs such as G.711, low−bit−rate codecs (such as G.729 and G.723.1) are highly optimized for voice patterns and tend to distort DTMF tones. This approach can result in problems accessing interactive voice response (IVR) systems. The **dtmf relay** command solves the problem of DTMF distortion by transporting DTMF tones "out of band" or separate from the encoded voice stream. If low bit−rate codecs (G.729, G.723) are used, turn on **dtmf relay** under the VoIP dial−peer.

- A typical conversation may contain 35−50% silence. Using Voice Activity Detection (VAD), silence packets are suppressed. For VoIP bandwidth planning, assume VAD will reduce bandwidth by 35%.

VAD is configured by default under the VoIP dial−peers. To enable/disable VAD, use the commands **router(config−dial−peer)#vad** and **router(config−dial−peer)# no vad** under the desired voip dial−peers.

Hardware and Software Requirements

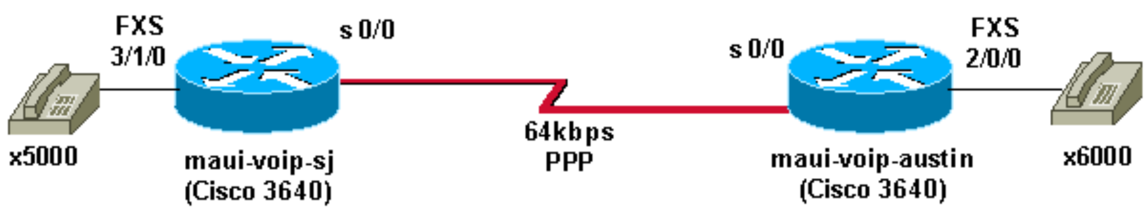
The information in this document was derived from an isolated lab environment. Ensure that you understand the potential impact of the commands on your network before using them.

These configurations were tested with the following equipment:

- Two Cisco 3640s with Cisco IOS® Software Release 12.1.2 (Enterprise Plus)

IP RTP Priority was introduced in Cisco IOS Release 12.0(5)T. LFI was introduced in Cisco IOS Release 11.3. Cisco IOS releases beyond 12.0.5T contain significant performance improvements for cRTP.

Network Diagram



Configurations

maui−voip−sj (Cisco 3640)

```
version 12.1
service timestamps debug datetime msec
service timestamps log uptime
service password−encryption
!
hostname maui−voip−sj
!
no logging console
aaa new−model
aaa authentication login default local
enable secret 5 <omitted>
!
username admin password 7 <omitted>
!

ip subnet−zero
no ip domain−lookup
!

voice−port 3/0/0
!
voice−port 3/0/1
!
voice−port 3/1/0
!
```

```
voice-port 3/1/1
!
dial-peer voice 1 pots
 destination-pattern 5000
 port 3/1/0
!-- <--More on Dial Peers
!
dial-peer voice 2 voip
 destination-pattern 6000
 session target ipv4:172.22.130.2
!
!-- If multiple physical interfaces are used, then keep the multilink interface
!-- without an IP address and add IP addresses to each physical interface.
interface Multilink1
 bandwidth 64
 !-- This command needs to be set correctly for the right fragment
 !-- size to be calculated.
 ip address 172.22.130.1 255.255.255.252
 ip tcp header-compression iphc-format
 no ip mroute-cache
 fair-queue
 no cdp enable
 ppp multilink
 ppp multilink fragment-delay 20
 !-- The configured value of 20 sets the fragment size such that all
 !-- fragments will have a 20ms maximum serialization delay.
 ppp multilink interleave
 multilink-group 1
 !-- This command links the multilink interface to the physical serial
 !-- interface.
 ip rtp header-compression iphc-format
 ip rtp priority 16384 16383 45
 !-- Assigns a priority queue for voice (based on UDP ports) with max
 !-- bandwidth = 45 Kbps.
!
interface Ethernet0/0
 ip address 172.22.113.3 255.255.255.0
!
interface Serial0/0
 bandwidth 64
 no ip address
 encapsulation ppp
 no fair-queue
 clockrate 64000
 ppp multilink
 multilink-group 1
 !-- This command links the multilink interface to the physical serial
 !-- interface.
!
router eigrp 69
 network 172.22.0.0
 no auto-summary
 no eigrp log-neighbor-changes
!
ip classless
no ip http server
!
line con 0
 transport input none
line 65 70
line aux 0
line vty 0 4
```

```
!
end
```

maui−voip−austin (Cisco 3640)

```
version 12.1
service timestamps debug datetime msec
service timestamps log datetime msec
service password−encryption
!
hostname maui−voip−austin
!
no logging console
aaa new−model
aaa authentication login default local
aaa authentication login NO_AUTHEN none
enable secret 5 <omitted>
!
username admin password 7 <omitted>
!
ip subnet−zero
!
voice−port 2/0/0
!
voice−port 2/0/1
!
voice−port 2/1/0
!
voice−port 2/1/1
!
dial−peer voice 1 pots
 destination−pattern 6000
 port 2/0/0
!
dial−peer voice 2 voip
 destination−pattern 5000
 session target ipv4:172.22.130.1
!
interface Multilink1
 bandwidth 64
 ip address 172.22.130.2 255.255.255.252
 ip tcp header−compression
 fair−queue
 no cdp enable
 ppp multilink
 ppp multilink fragment−delay 20
 ppp multilink interleave
 multilink−group 1
 ip rtp header−compression iphc−format
 ip rtp priority 16384 16383 45
!
interface Ethernet0/0
 ip address 172.22.112.3 255.255.255.0
!
interface Serial0/0
 no ip address
 encapsulation ppp
 no fair−queue
 ppp multilink
 multilink−group 1
!
router eigrp 69
 network 172.22.0.0
```

```
 no auto-summary
!
ip classless
no ip http server
!
line con 0
 login authentication NO_AUTHEN
 transport input none
line 33 40
 transport input telnet
line aux 0
line vty 0 4
!
end
```

Verification and Troubleshooting Commands

Before attempting any debug commands, please see Important Information on Debug Commands.For more info on the commands below, refer to the **Sample show and debug Output** section of this document.

- **show ip rtp header−compression** Displays RTP header compression statistics.
- **show interface [serial | multilink]** Use this command to check that status of the serial interface. Make sure the serial and multilink interface are up and open.
- **show ppp multilink** –This command displays bundle information for the Multilink PPP bundles.
- **debug priority** This debug command displays priority queueing events.
- **debug ppp multilink fragments** This debug command displays information about individual multilink fragments and interleaving events. This command output also identifies the sequence number of the packet and the fragment sizes.

- Troubleshooting Serial Line Problems
- Troubleshoot & Debug VoIP Calls − the Basics
- VoIP Debug Commands

Sample show and debug Output

```
!-- Interface Verification
!-- Make sure you see the following:
!-- LCP Open, multilink Open: Link control protocol (LCP) open statement
!-- indicates that the connection is establish.
!-- Open:IPCP. Indicates that IP traffic can be transmitted via the PPP link.
maui−voip−sj#show interfaces multilink 1
Multilink1 is up, line protocol is up
  Hardware is multilink group interface
  Internet address is 172.22.130.1/30
  MTU 1500 bytes, BW 64 Kbit, DLY 100000 usec,
     reliability 255/255, txload 27/255, rxload 1/255
  Encapsulation PPP, loopback not set
  Keepalive set (10 sec)
  DTR is pulsed for 2 seconds on reset
  LCP Open, multilink Open
```

```
  Open: IPCP
  Last input 00:00:03, output never, output hang never
  Last clearing of "show interface" counters 6d00h
  Input queue: 0/75/0/0 (size/max/drops/flushes); Total output drops: 0
  Queueing strategy: weighted fair
  Output queue: 0/1000/64/0/2441 (size/max total/threshold/drops/interleaves)
     Conversations  0/7/16 (active/max active/max total)
     Reserved Conversations 0/0 (allocated/max allocated)
  5 minute input rate 0 bits/sec, 0 packets/sec
  5 minute output rate 7000 bits/sec, 6 packets/sec
     115801 packets input, 7418654 bytes, 0 no buffer
     Received 0 broadcasts, 0 runts, 0 giants, 0 throttles
     0 input errors, 0 CRC, 0 frame, 0 overrun, 0 ignored, 0 abort
     354254 packets output, 35426589 bytes, 0 underruns
     0 output errors, 0 collisions, 0 interface resets
     0 output buffer failures, 0 output buffers swapped out
     0 carrier transitions

----------------------------------------------------------------

maui-voip-sj#show interface serial 0/0
Serial0/0 is up, line protocol is up
  Hardware is QUICC Serial
  MTU 1500 bytes, BW 64 Kbit, DLY 20000 usec,
     reliability 255/255, txload 1/255, rxload 1/255
  Encapsulation PPP, loopback not set
  Keepalive set (10 sec)
  LCP Open, multilink Open
  Last input 00:00:03, output 00:00:03, output hang never
  Last clearing of "show interface" counters 6d00h
  Queueing strategy: fifo
  Output queue 0/40, 0 drops; input queue 0/75, 0 drops
  5 minute input rate 0 bits/sec, 0 packets/sec
  5 minute output rate 0 bits/sec, 0 packets/sec
     220393 packets input, 9819949 bytes, 0 no buffer
     Received 0 broadcasts, 0 runts, 0 giants, 0 throttles
     8 input errors, 0 CRC, 8 frame, 0 overrun, 0 ignored, 0 abort
     553241 packets output, 55087341 bytes, 0 underruns
     0 output errors, 0 collisions, 16 interface resets
     0 output buffer failures, 0 output buffers swapped out
     34 carrier transitions
     DCD=up  DSR=up  DTR=up  RTS=up  CTS=up

------------------------------------------------------------

!-- Multilink & Fragment Size Verification
maui-voip-sj#show ppp multilink
Multilink1, bundle name is maui-voip-austin
  0 lost fragments, 0 reordered, 0 unassigned
  0 discarded, 0 lost received, 115/255 load
  0x14D received sequence, 0x186A sent sequence
  Member links: 1 active, 0 inactive (max not set, min not set)
    Serial0/0  160 weight
  !-- Notice the fragmentation size is 160 bytes. We configure the link with a
  !-- bandwidth of 64kbps and a serialization delay of 20ms.
  !-- Fragment Size (in bits) = bandwidth * serialization delay.

-------------------------------------------------------------------

!-- Debug priority command provides immediate feedback in case of VoIP packet
!-- drops or more than 5 packet are buffered in the priority queue.
!-- The output below shows the error message when VoIP packets are being dropped.
```

```
maui-voip-sj#debug priority
priority output queueing debugging is on
maui-voip-sj#
Mar 17 19:47:09.947: WFQ: dropping a packet from the priority queue 0
Mar 17 19:47:09.967: WFQ: dropping a packet from the priority queue 0
Mar 17 19:47:09.987: WFQ: dropping a packet from the priority queue 0

----------------------------------------------------------------------

!-- Testing Multilink PPP Link LFI
!-- The following output displays fragmentation and interleaving information
!-- when the the 64k PPP link is loaded with big data and VoIP packets.
maui-voip-sj#debug ppp multilink fragments
Multilink fragments debugging is on

Mar 17 20:03:08.995: Se0/0 MLP-FS: I seq C0004264 size 70
Mar 17 20:03:09.015: Se0/0 MLP-FS: I seq 80004265 size 160
Mar 17 20:03:09.035: Se0/0 MLP-FS: I seq 4266 size 160
Mar 17 20:03:09.075: Se0/0 MLP-FS: I seq 4267 size 160
Mar 17 20:03:09.079: Se0/0 MLP-FS: I seq 40004268 size 54
Mar 17 20:03:09.091: Se0/0 MLP-FS: I seq C0004269 size 70
Mar 17 20:03:09.099: Se0/0 MLP-FS: I seq C000426A size 70
Mar 17 20:03:09.103: Mu1 MLP: Packet interleaved from queue 24
Mar 17 20:03:09.107: Se0/0 MLP-FS: I seq C000426B size 70
Mar 17 20:03:09.119: Se0/0 MLP-FS: I seq C000426C size 70
Mar 17 20:03:09.123: Mu1 MLP: Packet interleaved from queue 24
Mar 17 20:03:09.131: Mu1 MLP: Packet interleaved from queue 24
Mar 17 20:03:09.135: Se0/0 MLP-FS: I seq C000426D size 70
Mar 17 20:03:09.155: Se0/0 MLP-FS: I seq C000426E size 70

-------------------------------------------------------------------

!-- Sample output of show ip rtp header-compression command
maui-voip-austin#show ip rtp header-compression
RTP/UDP/IP header compression statistics:
  Interface Multilink1:
    Rcvd:    1 total, 0 compressed, 0 errors
             0 dropped, 0 buffer copies, 0 buffer failures
    Sent:    0 total, 0 compressed,
             0 bytes saved, 0 bytes sent
    Connect: 16 rx slots, 16 tx slots, 0 long searches, 0 misses

----------------------------------------------------------------------

!-- This command displays information of the voip dial-peers.
maui-voip-sj#show dial-peer voice 2
VoiceOverIpPeer2
        information type = voice,
        tag = 2, destination-pattern = `6000',
        answer-address = `', preference=0,
        group = 2, Admin state is up, Operation state is up,
        incoming called-number = `', connections/maximum = 0/unlimited,
        application associated:
        type = voip, session-tMarget = `ipv4:172.22.130.2',
        technology prefix:
        ip precedence = 0, UDP checksum = disabled,
        session-protocol = cisco, req-qos = best-effort,
        acc-qos = best-effort,
        fax-rate = voice,   payload size =  20 bytes
        codec = g729r8,   payload size =  20 bytes,
        Expect factor = 10, Icpif = 30,signaling-type = cas,
        VAD = enabled, Poor QOV Trap = disabled,
```

```
        Connect Time = 283, Charged Units = 0,
        Successful Calls = 1, Failed Calls = 0,
        Accepted Calls = 1, Refused Calls = 0,
        Last Disconnect Cause is "10  ",
        Last Disconnect Text is "normal call clearing.",
        Last Setup Time = 93793451.
```

Related Information

- Configuring PPP and Multilink PPP
- Voice over IP Commands
- VoIP Debug Commands
- Configuring Voice over IP
- Voice over IP − Per Call Bandwidth Consumption
- Voice, Telephony and Messaging Technical Tips
- Voice and Telephony Technology Support Pages
- Voice and Telephony Product Support Pages